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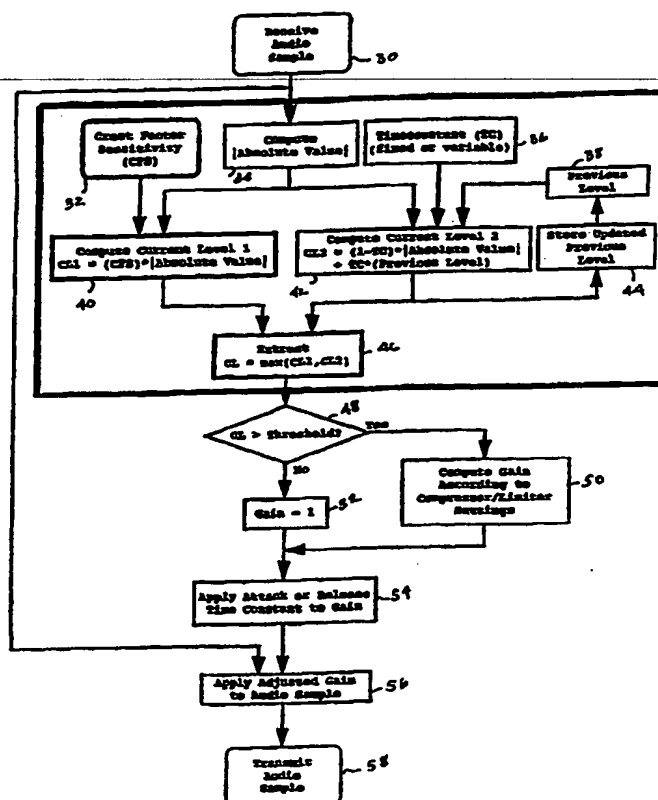
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(54) Title: AUDIO DYNAMIC RANGE PROCESSOR WITH ADJUSTABLE CREST FACTOR SENSITIVITY

(57) Abstract

The present invention provides an audio dynamic range signal processor. In one embodiment the processor uses an averaging detector (42) for receiving an input signal (30) and for generating a time–averaged value (42) of the input signal defined as the average signal magnitude. The processor includes a peak detector (40) for generating a value representative of the peak magnitude of the input signal defined as the peak signal magnitude, and an adjustable Crest Factor Sensitivity factor (32) for adjusting the peak signal magnitude to produce an adjusted peak signal magnitude. In addition, the processor has a selector (46) for setting a selected signal magnitude as the larger of the average signal magnitude and the adjusted peak signal magnitude. The processor has a dynamic processor for adjusting the amplitude of the input signal with the variable gain (56), where the dynamic processor gain is dependent upon the selected signal magnitude.

AUDIO DYNAMIC RANGE PROCESSOR WITH
ADJUSTABLE CREST FACTOR SENSITIVITY

BACKGROUND OF THE INVENTION

5   The present invention relates to an audio processing system and more particularly to an audio dynamic range processor with adjustable crest factor sensitivity.

      Prior art audio dynamic range compressors and limiters accommodate signals with varying crest factors (ratio of peak magnitude to average magnitude) by means of

10   an "observation window" (period of time over which the magnitude of the input signal is computed) which is selectable between "average" and "peak". Choice of the appropriate "observation window" is always a major compromise between the fairly long interval associated with the "average" window and the extremely short interval associated with the "peak" window.

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SUMMARY OF THE INVENTION

      The present invention provides an audio dynamic range signal processor. In one embodiment the processor uses an averaging detector for receiving an input signal and for generating a time-averaged value of the input signal defined as the average

20   signal magnitude. The processor includes a peak detector for generating a value representative of the peak magnitude of the input signal defined as the peak signal magnitude, and an adjustable Crest Factor Sensitivity factor for adjusting the peak signal magnitude to produce an adjusted peak signal magnitude. In addition, the processor has a selector for setting a selected signal magnitude as the larger of the

25   average signal magnitude and the adjusted peak signal magnitude. The processor has a dynamic processor for adjusting the amplitude of the input signal with variable gain, where the dynamic processor gain is dependent upon the selected signal magnitude.

DESCRIPTION OF THE DRAWINGS

30   The present invention is readily understood with reference to the following specification and drawing wherein:

FIG. 1 is a block diagram of the present invention; and

FIG. 2 is a software flow diagram of the present invention.

DETAILED DESCRIPTION

5   Figure 1 shows a block diagram of the components of a preferred embodiment of the audio signal processor with adjustable crest factor sensitivity. The preferred embodiment uses digital signal processing. However, the invention may also be implemented in the analog domain.

Referring first to FIG. 1, analog audio input signals designated by the arrow 20

10   are applied to an analog to digital converter (ADC) 22, for example a Crystal Model No. CS5390. The input signal may instead be some component of a video signal. The digitized audio signals from the ADC 20 are then applied to a digital signal processing (DSP) computer 24, for example a Motorola Model No. DSP 56004 and processed as set forth in FIG. 2 and described below. The output of the DSP 24 is

15   then applied to a digital to analog converter (DAC), for example, a Crystal Model CS 4329, and converted back to analog signals, as represented by the arrow 28.

The software for the DSP 24 is illustrated in FIG. 2. In step 30, a single sample of audio data is received from the A/D converter 22. The Crest Factor Sensitivity "CFS" parameter is selected by an operator in step 32, based upon

20   dynamics of audio signal and listening preferences. Values of the CFS range from 0.0 through 1.0; at 0.0 signal peaks are ignored and only average values are used to compute signal level, at 1.0 the signal peak is left unweighted.

The crest factor is defined as the ratio of the peak value of a signal to its average. In this representative embodiment, the level detector will respond to the

25   average signal level unless a peak is encountered whose crest factor exceeds the reciprocal of the CFS parameter. The CFS parameter thus represents the reciprocal of the crest factor that a signal must possess before the detector responds to peaks rather than averages.

In step 34, the absolute value of the audio sample received in step 30 is

30   computed by the DSP 24. The exponential time constant "TC," as used in this embodiment, is computed and stored in computer memory in step 36. Selection of a proper time constant is well-known in the art. Subsequently in step 38, the Current

Level 2 "CL2" parameter that was calculated for the previous audio sample is stored in computer memory.

A Current Level 1 "CL1" parameter is computed as the product of the CFS parameter, which ranges from 0.0 through 1.0, and the absolute value of the current audio sample in step 40 (i.e., the peak value). The value CL1 thus represents the weighted or adjusted instantaneous amplitude of the input signal. A Current Level 2 "CL2" parameter is calculated in step 42 for the current audio sample using the TC parameter obtained from step 36, the absolute value of the current audio sample computed in step 34, and the previous value of the CL2 parameter stored in step 38. This algorithm implements a first-order lowpass filter through which the absolute values of the audio data are passed, thus representing the averaged level, or average signal magnitude, of the audio data. Steps 34 - 38, and 42 - 44 are therefore a detection of the average signal magnitude.

Such first-order lowpass filters algorithms as used by the average detector in steps 34 - 38, and 42 - 44 are well-known in the art. Many types may be used, for example, chapter 6.4 of the textbook, A. Oppenheim, A. Willsky, and I. Young *Signals and Systems* (1983, Prentice-Hall, Inc.), provides different types of lowpass filter algorithms that include a time constant, such as TC.

The CL2 parameter for the current audio sample is stored in computer memory to be used as the "Previous Level" parameter for the subsequent audio sample in step 44. Thus, the average detector regenerates the CL2 parameter after the reception of each audio sample.

The Current Level "CL" parameter is selected in step 46. The CL parameter is set as the larger of the CL1 and CL2 parameters. In this way the average level CL2 is defined as the selected signal magnitude, CL, unless a peak in the audio data is encountered that is large enough to overcome the weighting.

The selected signal magnitude calculated in step 46 controls the processing of the input signal performed by a dynamic processor of the DSP 24. The dynamic processor is used for adjusting the amplitude or signal level of the input signal with a variable gain. The dynamic processor's variable gain is dependent upon the selected signal magnitude; thus, the selected signal magnitude controls the dynamic processor's processing by setting the dynamic processor's gain.

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The dynamic processor described above, like typical dynamic processors known in the art, may be comprised of a compressor, a limiter, a gate, an expander, a ducker, a de-esser, or any combination thereof. As is known in the art (see, e.g., B. Hurtig *Multi-Track Recording for Musicians* at pp. 55-59 (1988, Alfred Publishing
5   Co.)), dynamic processors control the amplitude levels, or dynamic range, of audio signals. Dynamic processors, such as compressors, limiters, de-essers, and duckers, *reduce* the dynamic range of audio signals. Conversely, dynamic processors, such as expanders and gates, *increase* the dynamic range of audio signals.

For instance, compressors reduce the level of signals which pass a threshold
10   value, while limiters prevent the level of signals from exceeding the threshold level. Expanders, on the other hand, reduce the level of signals which fall below a threshold value, while gates virtually silence signals that fall below the threshold level. Duckers typically compress, or suppress, a second signal under the amplitude of the input signal when the input signal magnitude exceeds a threshold level. De-essers, as their
15   name indicates, are designed to reduce the "hiss" associated with sibilance. De-essers accomplish this by compressing the appropriate frequencies (approximately 2000 - 6000 Hz) of an input signal associated with sibilance ("sibilance frequencies"), when the signal level in the sibilance frequencies of the input signal exceeds a threshold level.

20   Past dynamic compressors, however, have not used the input signal's magnitude, calculated with an adjustment for its crest factor as described above, as the basis for comparison against the threshold level.

The embodiment shown in steps 48 - 56 in Figure 2 employs a compressor as the dynamic processor. In step 48, the amplitude of the audio data, represented by the
25   CL parameter, is compared with the previously established threshold in step 48. If the amplitude is above the threshold then control passes to step 50. If the amplitude is less than or equal to the threshold then control passes to step 52.

In step 50, the amplitude of the audio data is determined to be above the threshold. The value of the Gain parameter (which is actually an attenuation in this
30   case; "gain" is used here in its generic sense so that it can represent either boost or cut) is determined such that it will reduce the amplitude of the audio data to the

appropriate level based upon the previously established compression ratio settings. Methods for determining the gain value are well known in the art.

In step 52, the amplitude of the audio data is determined to be below the threshold. Signals with an amplitude below the threshold are unaffected by the 5 compressor, therefore the Gain parameter is set to 1.0.

Sudden changes in the value of the Gain parameter are prevented through the use of "Attack time" (in the case of a reduction in Gain) or "Release Time" (in the case of an increase in Gain) implementations that ramp the gain value gradually from its previous value to its new value in step 54. Methods for implementing Attack and 10 Release Times are well-known in the art.

The adjusted value of the current audio sample is computed as the product of the Gain value determined in step 56 and the audio sample obtained from step 30. The adjusted audio sample is sent to the D/A converter 26 in step 58.

In operation, the preferred embodiment solves the problems described above 15 associated with past signal processors. The preferred embodiment effectively employs two "observation windows" (effective window-of-time over which the signal magnitude is computed) -- one of length suitable for "average" magnitude computations and the other of length factor, continuously adjustable between 0.0 and 1.0, is applied to the magnitude computed by the "peak" detector as a means of 20 controlling sensitivity to crest factor. At each point in time the "average" magnitude is compared with weighted "peak" magnitude and the larger of the two is considered to be the signal magnitude for use in subsequent operations of the compressor (dynamic processor).

Use of the "weighting" factor as a crest factor sensitivity adjustment is 25 demonstrated with the following examples, in which it is assumed for convenience that the "average" detector performs a true RMS computation:

The crest factor of a sinewave is $\sqrt{2}$. If the "weighting" factor applied to the "peak" detector is set to $1/\sqrt{2}$, then the "average" detector and the "peak" detector will compute exactly the same magnitude for the sinewave. If, however, the 30 "weighting" factor is set to 1.0, then the "peak" detector will compute a magnitude that is $\sqrt{2}$ times the magnitude computed by the "average" detector for the same

sinewave signal, and the compressor attenuation will be set according to this higher value.

If the input signal consists of a sinewave of RMS amplitude 1.0 combined with a very short impulse of amplitude 10.0, then the "average" detector will compute a magnitude of slightly greater than 1.0 while the "peak" detector will compute 10.0. By setting the "weighting" factor to less than 1/10, the magnitude computed by the "peak" detector will be ignored in favor of the magnitude computed by the "average" detector. With any "weighting" factor greater than 1/10, though, the weighted peak value will be larger than the average value and will thus be considered by the detector to be the signal magnitude.

In the manner described above, the magnitude computation may be set to ignore peaks altogether in favor of average values ("weighting" factor = 0.0), or to ignore average values altogether (unless the signal segment is constant) in favor of peak values ("weighting" factor = 1.0), or to anything in between.

While a preferred embodiment of the present invention has been described, it should be understood that various changes, adaptations and modifications may be made therein without departing from the spirit of the invention and the scope of the appended claims.

WHAT IS CLAIMED IS:

1.   An audio dynamic range signal processor, comprising:

an averaging detector for receiving an input signal and for generating a time-averaged value of the input signal defined as the average signal magnitude;

a peak detector for generating a value representative of the peak magnitude of the input signal defined as the peak signal magnitude;

an adjustable Crest Factor Sensitivity factor for adjusting the peak signal magnitude to produce an adjusted peak signal magnitude; and

a selector for setting a selected signal magnitude as the larger of the average signal magnitude and the adjusted peak signal magnitude; and

a compressor for compressing the input signal when the selected signal magnitude exceeds a predetermined threshold level, the compressor having variable gain dependent upon the selected signal magnitude.

2.   The signal processor of claim 1, wherein the time-average value of the input signal generated by the averaging detector is the root-mean-square average value of the input signal.

3.   The signal processor of claim 1, wherein the averaging detector comprises a lowpass filter algorithm receiving the squared-magnitude of the input signal.

4.   The signal processor of claim 1, wherein the averaging detector comprises a lowpass filter algorithm receiving the absolute value of the input signal.

5.   The signal processor of claim 1, wherein the adjusted peak signal magnitude is computed as the product of the peak signal magnitude and the Crest Factor Sensitivity factor.

6.   The signal processor of claim 5, wherein the Crest Factor Sensitivity factor is continuously adjustable between 0.0 and 1.0.

7.   The signal processor of claim 6, wherein the Crest Factor Sensitivity factor is manually adjustable.

8.   The signal processor of claim 1, wherein the compressor gain is unity when the selected signal magnitude does not exceed the predetermined threshold level.

9. The signal processor of claim 1, wherein the selector sets the selected signal magnitude as the average signal magnitude unless the Crest Factor of the input signal exceeds the reciprocal of the Crest Factor Sensitivity factor.

10. The signal processor of claim 1, wherein the input signal comprises a series of discrete samples of an analog waveform.

11. The signal processor of claim 1, wherein the input signal comprises a digital signal.

12. The signal processor of claim 10, wherein the peak signal magnitude is regenerated with each discrete sample.

13. The signal processor of claim 12, wherein the peak signal magnitude is the magnitude of each discrete sample.

14. The signal processor of claim 10, wherein the average signal magnitude is regenerated with each discrete sample.

15. The signal processor of claim 14, wherein the averaging detector comprises a lowpass filter algorithm receiving the squared-magnitude of each discrete sample.

16. The signal processor of claim 14, wherein the averaging detector comprises a lowpass filter algorithm receiving the absolute value of each discrete sample.

17. A method of processing an audio dynamic range signal, comprising:

receiving an input signal;

generating a time-average value of the input signal defined as the average signal magnitude;

generating a value representative of the peak magnitude of the input signal defined as the peak signal magnitude;

adjusting the peak signal magnitude by an adjustable Crest Factor Sensitivity factor to produce an adjusted peak signal magnitude;

setting a selected signal magnitude as the larger of the average signal magnitude and the adjusted peak signal magnitude; and

adjusting the amplitude of the input signal with variable gain, the gain being dependent upon the selected signal magnitude.

18. An audio dynamic range signal processor, comprising:

an averaging detector for receiving an input signal and for generating a time-averaged value of the input signal defined as the average signal magnitude;

a peak detector for generating a value representative of the peak magnitude of
the input signal defined as the peak signal magnitude;

an adjustable Crest Factor Sensitivity factor for adjusting the peak signal
magnitude to produce an adjusted peak signal magnitude; and

5   a selector for setting a selected signal magnitude as the larger of the average
signal magnitude and the adjusted peak signal magnitude; and

a dynamic processor for adjusting the amplitude of the input signal with
variable gain, the dynamic processor gain being dependent upon the selected signal
magnitude.

10   19.   The signal processor of claim 18, wherein the dynamic processor comprises a
compressor, the compressor for compressing the input signal when the selected signal
magnitude exceeds a predetermined threshold level, the compressor having variable
gain dependent upon the selected signal magnitude.

20.   The signal processor of claim 18, wherein the dynamic processor comprises a
15   limiter, the limiter for limiting the amplitude of the input signal when the selected
signal magnitude exceeds a predetermined threshold level.

21.   The signal processor of claim 18, wherein the dynamic processor comprises an
expander, the expander for expanding the amplitude of the input signal when the
selected signal magnitude falls below a predetermined threshold level, the expander
20   having variable gain dependent upon the selected signal magnitude.

22.   The signal processor of claim 18, wherein the dynamic processor comprises a
ducker, the ducker for compressing the amplitude of a second signal under the
amplitude of the input signal when the selected signal magnitude exceeds a
predetermined threshold level, the ducker having variable gain dependent upon the
25   selected signal magnitude.

23.   The signal processor of claim 18, wherein the dynamic processor comprises a
gate, the gate for squelching the input signal when the selected signal magnitude falls
below a predetermined threshold level.

24.   A signal processor, comprising:
30   an averaging detector for receiving an input signal and for generating a time-
averaged value of the input signal defined as the average signal magnitude;

a peak detector for generating a value representative of the peak magnitude of the input signal defined as the peak signal magnitude;

an adjustable Crest Factor Sensitivity factor for adjusting the peak signal magnitude to produce an adjusted peak signal magnitude; and

5   a selector for setting a selected signal magnitude as the larger of the average signal magnitude and the adjusted peak signal magnitude; and

a dynamic processor for adjusting the amplitude of the input signal with variable gain, the dynamic processor gain being dependent upon the selected signal magnitude.
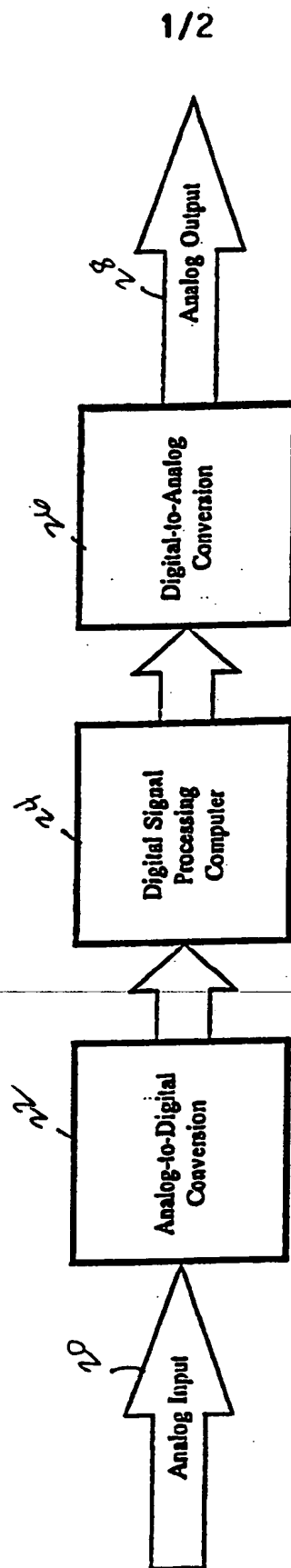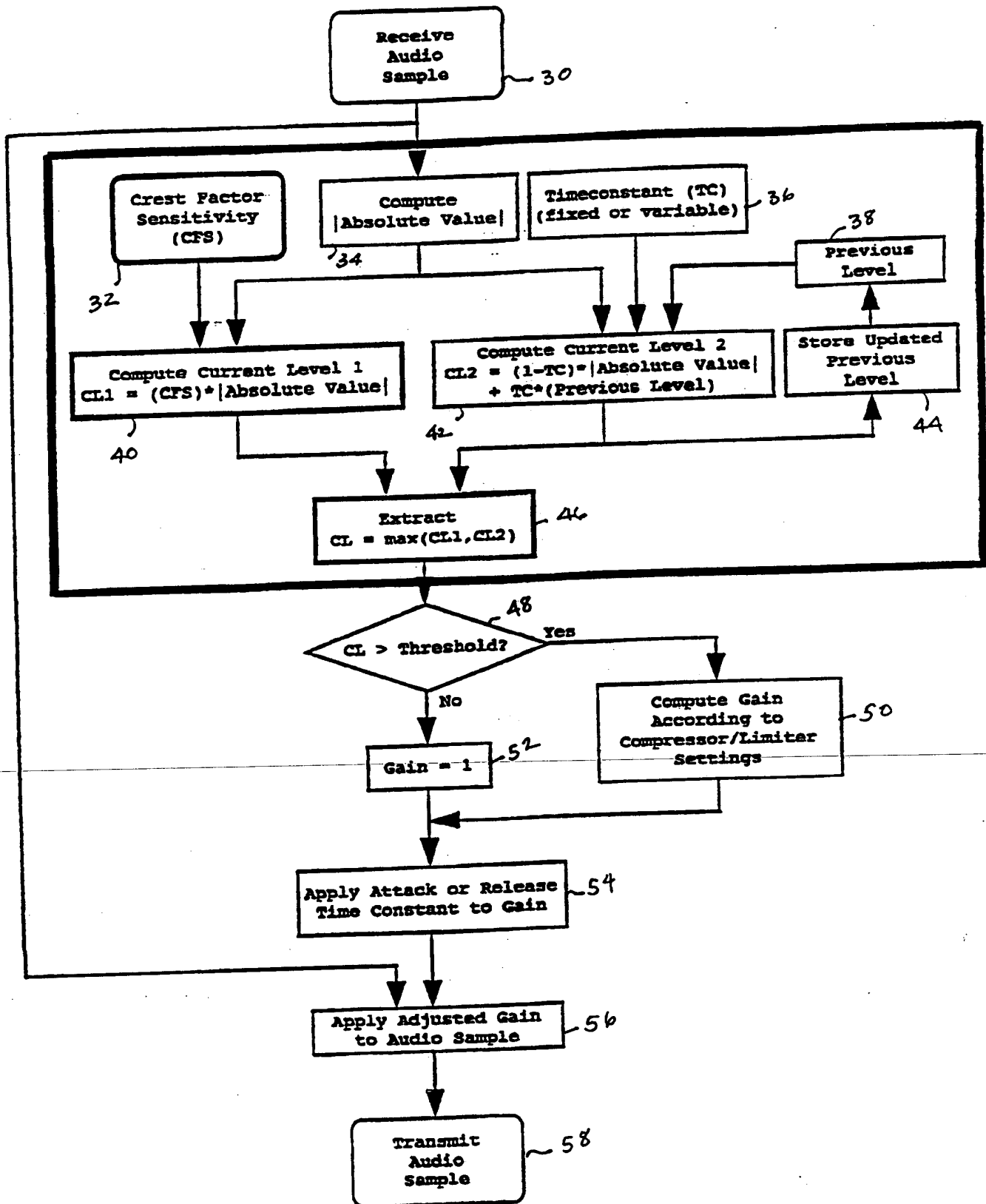
FIG 1.

Analog Output

28

Digital-to-Analog
Conversion

20

Digital Signal
Processing
Computer

24

Analog-to-Digital
Conversion

22

Analog Input

26

Receive
Audio
Sample ⟶ 30

Crest Factor
Sensitivity
(CFS) — 32

Compute
|Absolute Value| — 34

Timeconstant (TC)
(fixed or variable) — 36

Previous
Level — 38

Compute Current Level 1
CL1 = (CFS)*|Absolute Value| — 40

Compute Current Level 2
CL2 = (1-TC)*|Absolute Value|
+ TC*(Previous Level) — 42

Store Updated
Previous
Level — 44

Extract
CL = max(CL1,CL2) — 46

CL > Threshold? — 48

Yes

No

Compute Gain
According to
Compressor/Limiter
Settings — 50

Gain = 1 — 52

Apply Attack or Release
Time Constant to Gain — 54

Apply Adjusted Gain
to Audio Sample — 56

Transmit
Audio
Sample — 58

FIG 2.

A.   CLASSIFICATION OF SUBJECT MATTER

IPC(6)   :H03G 7/00

US CL   :381/106.107; 333/14

According to International Patent Classification (IPC) or to both national classification and IPC

B.   FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. :   381/106.107; 333/14

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

None

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

None

C.   DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 4,250,471 A (DUNCAN) 10 February 1981 (10.02.81), see Figures 9-11 and 13, and column 15, line 30 to column 17, line 37.	1-21,23, and 24
A	US 5,255,325 A (ISHIMITSU ET AL) 19 October 1993 (19.10.93).	None
A	US 5,267,322 A (SMITH ET AL) 30 November 1993 (30.11.93).	None

☐   Further documents are listed in the continuation of Box C.   ☐   See patent family annex.

*	Special categories of cited documents:	"T"	later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"A"	document defining the general state of the art which is not considered to be of particular relevance	"X"	document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"E"	earlier document published on or after the international filing date		
"L"	document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	"Y"	document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"O"	document referring to an oral disclosure, use, exhibition or other means		
"P"	document published prior to the international filing date but later than the priority date claimed	"&"	document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
20 JUNE 1998	2 7 AUG 1998

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